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SPECIFICATION

TITLE OF THE INVENTION

METHOD AND APPARATUS FOR CONNECTING COMMUNICATION DEVICE
VIA IP NETWORK

5 BACKGROUND OF THE INVENTION

 This invention relates to a method and apparatus
for connecting a communication device such as a
telephone via an IP (Internet Protocol) network. More
particularly, the invention relates to a communication
10 device connecting method and apparatus in which when a
call originate request has been issued to make a
connection to a called terminal via an IP network, the
originating terminal is connected to the terminating
terminal upon taking into account the communication
15 quality of the route through the IP network.

 IP telephone systems for placing telephone calls
via an IP network currently have been spotlighted as
telephone systems for the next generation to supplant
existing telephone systems based upon STM (Synchronous
20 Transfer Mode) networks. Arranging it so that ease of
use and convenience on a par with that of existing
telephone schemes can be realized is an important
challenge confronting IP telephony. Various modes of
utilizing conventional IP telephone systems have been
25 proposed and implemented. One such example is an
Internet relay telephone service provided to general
users such as individuals by an Internet telephone
service provider. The purpose of this system is to

lower the cost of calls by using the Internet as an intermediary.

Fig. 9A is a diagram useful in describing such an Internet relay telephone service. When the user of a telephone 1 dials up an access point of an Internet telephone service provider, an IP packetizing unit 3 responds via an exchange (STM telephone network) 2 and allows the user to enter the telephone number of the call destination. On the basis of the entered telephone number, the IP packetizing unit 3 queries a server 5, referred to as a "gatekeeper", via an IP network 4 with regard to the IP address of an IP packetizing unit 6 at the destination of transferred packets. Upon learning of the IP address, the IP packetizing unit 3 executes a procedure (stipulated by Recommendation H.225.0) for setting up an IP connection to the IP packetizing unit 6 of the called party. If the IP connection is established, then the telephone call is implemented via a route constituted by the originating telephone 1, the exchange 2, the IP packetizing unit 3, the IP network 4, the IP packetizing unit 6, an exchange 7 and a terminating telephone 8 in the order mentioned. It should be noted, however, that the transfer of actual voice data (voice packets) is performed between the IP packetizing units 3 and 6 using RTP (Real Time Protocol).

Thus, to place a telephone call from an originating telephone to a terminating telephone via an IP network

in accordance with the prior art, it will suffice to dial the IP-network access telephone number and the telephone number of the terminating telephone. To make a telephone call via an existing STM network, only the
5 telephone number of the terminating telephone need be dialed.

An IP network offers best-effort traffic service and uses the network band to the fullest extent possible; it does not compensate for bandwidth and
10 communication quality. As a consequence, if the IP network becomes congested, packet loss PL occurs, as shown in Fig. 9B, a packet delay PD is produced and delay time varies owing to the characteristic of the IP network. Thus, though the IP network offers the
15 advantage of low communication cost, there are instances where voice is delayed or interrupted by congestion, resulting in degraded voice quality. This differs from the case where use is made of an STM network.

The state of the art is such that if an IP network
20 has been accessed to make a telephone call, the call must be continued regardless of whether voice quality is good or bad. If voice quality is poor and the caller wishes to re-connect via an STM network, the user must take the trouble to first break the connection and then
25 redial.

SUMMARY OF THE INVENTION

Accordingly, an object of the present invention is to so arrange it that when a call originate request is

issued to make a connection to a terminating terminal
via an IP network, the call can be implemented with
excellent communication quality taking into account the
communication quality of the route through the IP
5 network.

Another object of the present invention is to so
arrange it that a telephone call is connected via a
route through an IP network if the communication quality
of the route is good, whereas if the quality through
10 this route is bad, the call is connected by selecting an
alternative route within the IP network or another route
through a network (e.g., an STM or ATM network) other
than the IP network, whereby it is possible to implement
the call with excellent communication quality.

15 A further object of the present invention is to so
arrange it that if the communication quality of a route
through an IP network is bad, a call is connected by
selecting an alternative route within the IP network or
another route through a network (e.g., an STM or ATM
20 network) other than the IP network in accordance with a
command from the calling party or automatically in
accordance with a determination made by the switching
system, whereby it is possible to implement the call
with excellent communication quality.

25 Yet another object of the present invention is to
so arrange it that another route can be selected by a
designation made by the caller through a simple
operation without requiring that the caller take the

trouble to disconnect and then redial.

In accordance with the present invention, the foregoing objects are attained by (1) evaluating voice quality along a route through an IP network from a switching system on an originating side to a switching system on a terminating side when a call originate request has been issued to connect an originating terminal to a terminating terminal via the IP network; (2) connecting the originating terminal and the terminating terminal if the communication quality is good; and (3) if the communication quality is bad, controlling connection of the communication device in accordance with a command from the caller or performing control by selecting a route other than the first-mentioned route, e.g., an alternative route within the IP network or another route via a network (STM or ATM network) other than the IP network, automatically in accordance with a determination made by a switching system. If this arrangement is adopted, a telephone call can be implemented with an excellent communication quality.

In the present invention, the evaluation of communication quality is performed by (1) sending a test packet from the switching system on the originating side to the switching system on the terminating side via the route through the IP network before a channel is established; (2) sending an ACK packet from the switching system of the terminating side to the

switching system on the originating side in response to receipt of the test packet; (3) measuring, at the exchange on the originating side, delay time from sending of the test packet to receipt of the ACK packet; 5 and (4) comparing length of the delay time with length of a set time. If this arrangement is adopted, control to connect a communication device can be performed upon measuring communication quality currently prevailing in the IP network. This makes it possible to achieve 10 excellent call quality in reliable fashion.

In the present invention, evaluation of communication quality can be performed in the following manner as well: (1) preserving packet loss rate, etc., that prevailed during a call after the call ends, and 15 (2) determining whether communication quality is good or bad by referring to the loss rate. If this arrangement is adopted, it is unnecessary to measure communication quality before the channel is established, as a result of which it is possible to shorten time required for a 20 call to start.

In the present invention, if communication quality of a route through the IP network is not good, the switching system reports this to the caller by communication and the caller, by operating keys on a 25 terminal, instructs the exchange to perform one of the following control operations: (1) to continue connection control via the above-mentioned route for which the communication quality has been determined to

be no good; (2) to perform connection control via an alternative route through the IP network; (3) to perform connection control through another route via a network (e.g., STM or ATM network) other than the IP network; and (4) to perform disconnect control. If this arrangement is adopted, the caller can specify control for the next communication device connection through a simple operation without being required to reconnect.

In the present invention, if communication quality of a route through the IP network is not good, a switching system automatically selects an alternative route through the IP network or a route which connects the originating terminal and the terminating terminal via a network (e.g., STM or ATM network) other than the IP network, and connects these terminals via the selected route. If this arrangement is adopted, greater convenience is provided because an excellent route can be selected without requiring that the caller perform any operation.

Other features and advantages of the present invention will be apparent from the following description taken in conjunction with the accompanying drawings, in which like reference characters designate the same or similar parts throughout the figures thereof.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a diagram useful in describing an overview of the present invention;

Fig. 2 is a block diagram illustrating the construction of a network to which the telephone connecting control method of the present invention is applied;

5 Fig. 3 is a diagram useful in describing subscriber data and system data;

Fig. 4 is a block diagram illustrating the construction of an IP packetizing unit according to the present invention;

10 Figs. 5A and 5B are diagrams useful in describing packet structures;

Fig. 6 is a diagram useful in describing the procedure of processing for controlling telephone connection in a case where a pre-call voice-quality measurement mode is adopted;

15

Fig. 7 is a status-transition diagram useful in describing overall telephone connection control;

Figs. 8A to 8J are flowcharts of a variety of processing; and

20 Figs. 9A and 9B are diagrams useful in describing a mode of using an IP telephone according to the prior art as well as the problems of the prior art.

DESCRIPTION OF THE PREFERRED EMBODIMENT

(A) Overview of the present invention

25 (a) Overall construction

Fig. 1 is a diagram useful in describing an overview of the present invention. Shown in Fig. 1 are telephones 10, 60, exchanges 20, 50, an IP network 30

and a network (e.g., an STM network) 40 other than an IP network. The exchange 20 includes an STM switch network (call channel circuit) 21, a central controller (CC) 22, a main memory (MM) 23 and an IP packetizing unit 24 for
5 executing processing in accordance with various protocols, IP-packetizing voice data, sending the voice-data packets to the IP network 30, receiving IP packets from the IP network 30, restoring the IP packets to voice data and outputting the voice data to the STM
10 switch network 21. If the voice quality of a route through the IP network is poor, an announcement unit 25 reports this fact to the calling party via the telephone 10. The exchange 50 includes an STM switch network 51, a central controller (CC) 52, a main memory (MM) 53 and
15 an IP packetizing unit 54.

(b) Overview of telephone connection via IP network

When a call originate request is issued from the originating telephone 10 to connect this telephone to
20 the terminating telephone 60 via the IP network 30, the central controller (CC) 22 of the exchange 20 on the originating side (1) evaluates the voice quality along the route through the IP network 30 from the IP packetizing unit 24 on the originating side to the IP
25 packetizing unit 54 on the terminating side. (2) If the voice quality is good, the central controller 22 connects the originating telephone 10 and the terminating telephone 60 via the above-mentioned route.

(3) If the voice quality is poor, then the central controller 22 performs telephone connecting control in accordance with a command issued by the caller from the telephone 10 or performs control upon selecting a route
5 other than the above-mentioned route, e.g., an alternative route within the IP network or another route via a network (STM or ATM network) other than the IP network, automatically in accordance with a determination made by the central controller 22
10 independently. If this arrangement is adopted, a telephone call can be implemented with an excellent voice quality.

(c) Methods of evaluating voice quality

(c-1) First evaluation method

15 In order to evaluate voice quality, (1) the central controller 22 instructs the IP packetizing unit 24 to send a test packet to the IP packetizing unit 54 on the terminating side before a channel is established. (2) The IP packetizing unit 24 responds by creating a test
20 packet 100 and sending the test packet 100 to the IP packetizing unit 54 on the terminating side via the IP network 30. (3) If the test packet 100 is received, the IP packetizing unit 54 creates an ACK packet 200 and sends the ACK packet 200 to the IP packetizing unit 24
25 on the transmitting side. (4) The central controller 22 measures the time ($\alpha + \beta$) that elapses from issuance of the test packet 100 to receipt of the ACK packet 200 and compares the length of this measured time with the

length of a set time to thereby determine evaluate voice quality. More specifically, if the measured time is shorter than the set time, this means that the delay is small. Hence a decision is rendered to the effect that voice quality is good. If the measured time is longer than the set time, the delay is large and therefore a decision is rendered to the effect that voice quality is poor. If this arrangement is adopted, the telephone connection can be controlled upon measuring voice quality in the IP network at the time of an outgoing call. This makes it possible to achieve excellent voice quality in reliable fashion.

(c-2) Second evaluation method

Voice quality can be evaluated in the manner described below as well. Specifically, (1) the central controller 22 calculates packet loss rate η ($= N_{PL}/N_{PT}$) from lost packet count N_{PL} and transmitted packet count N_{PT} prevailing during a call and reported by the IP packetizing unit 24, and preserves the calculated loss rate for each route after the call ends, and (2) the central controller 22 determines, before the channel is established, whether voice quality is good or bad by referring to the loss rate corresponding to the route that connects the originating telephone and the terminating telephone. If this arrangement is adopted, it is unnecessary to measure voice quality by sending and receiving the test packet and ACK packet before the channel is established, as a result of which it is

possible to shorten time required for a call to start.

(d) Control in accordance command from caller

If the voice quality of the route through the IP network is poor, the central controller 22 connects the announcement unit 25 and the originating telephone 10 via the STM switch network 21 and provides the caller with a voice announcement from the announcement unit 25 to the effect that the voice quality is poor. Using keys on the telephone 10, the caller instructs the central controller 22 to perform one of the following control operations: (1) to continue connection control via the above-mentioned route for which the voice quality has been determined to be no good; (2) to perform connection control via an alternative route through the IP network; (3) to perform connection control via another route via a network other than the IP network; and (4) to perform disconnect control. Thereafter, the central controller 22 exercises connection control in accordance with the command from the caller. As a result, even if the voice quality of the route through the IP network is poor, the caller can instruct the exchange to perform the next connection control operation through a simple manipulation without requiring redialing.

(e) Automatic control by system

If voice quality of a route through the IP network is poor, the central controller 22 can automatically select a route that connects the originating telephone

and the terminating telephone via a network (e.g., STM or ATM network) other than the IP network and can connect these terminals via the selected route. If this arrangement is adopted, greater convenience is provided
5 because a telephone conversation can be implemented with good voice quality without requiring that the caller perform any operation.

It should be noted that an arrangement is adopted in which both the control method (d) in accordance with
10 a command from the caller and the system automatic control method (e) are possible. In this case it is determined by agreement with the subscriber as to which method should be used for controlling the telephone connection. If the voice quality through the IP network
15 is poor, the telephone connection is controlled in accordance with the method that has been determined.

(B) Network configuration

(a) Overall configuration

Fig. 2 is a block diagram illustrating the
20 configuration of a network to which the telephone connection control method of the present invention is applied. Components shown in Fig. 2 identical with those of Fig. 1 are designated by like reference characters. Shown in Fig. 2 are the telephones 10, 60,
25 the exchanges 20, 50, the IP network 30, an IP network 31, the STM network 40, an ATM (Asynchronous Transfer Mode) network 41 and a server 70. By agreement made in advance, the telephone 10 is capable of being connected

via the IP network and via a network (e.g., the STM or ATM network) other than the IP network. The server 70 stores the correspondence between telephone numbers and IP addresses of IP packetizing units and, in response to
5 a request from the exchange on the originating side, provides the IP address of the IP packetizing unit on the terminating side that corresponds to the telephone number of the terminating telephone.

The front panel 20 includes the STM switch network
10 21, the central controller (CC) 22, the main memory (MM) 23, the IP packetizing unit 24, the announcement unit (ANM) 25, a cell assembler/disassembler 26 and a device (not shown) for generating various tones. The STM switch network 21 accommodates a subscriber line 27,
15 which is for accommodating the telephone 10, and a number of trunks such as an ATM trunk 28a, an IP trunk 28b and an ATM trunk 28c. The STM trunk, IP trunk and ATM trunk construct trunk groups (an STM trunk group, IP trunk group and ATM trunk group) at the respective
20 destinations, and the central controller 22 manages the state of use and state of congestion of the trunks in each trunk group using a management table.

The IP packetizing unit 24 executes processing in accordance with various protocols, converts voice data
25 to IP packets, sends the packet to the IP network 30, restores IP packets received from the IP network 30 to voice data and outputs the voice data to the STM switch network 21. The exchange 50 has a construction similar

to that of the exchange 20 and includes the STM switch network 51, the central controller (CC) 52, the main memory (MM) 53, the IP packetizing unit 54 and a cell assembler/disassembler 56.

5 (b) Subscriber data / system data

Subscriber data SBDT and system data SYDT is stored in the main memory 23 of the exchange 20 in the manner shown in Fig. 3. The subscriber data SBDT contains contract type 301 and call data 302 stored in
10 association with the telephone number of a subscriber. In a case where the voice quality of a route through an IP network is poor, the contract type 301 specifies whether the connection of the telephone is to be controlled in accordance with a command from the caller
15 or upon selecting the route automatically in accordance with a determination made by the system. In the case of the former, the caller operates keys on the originating telephone (enters a specific number) to instruct the central controller 22 to perform one of the following
20 control operations: (1) to continue connection control via the route for which the voice quality is poor; (2) to perform connection control via an alternative route through the IP network; (3) to perform connection control via another route via a network (STM or ATM
25 network) other than the IP network; and (4) to perform disconnect control. The central controller 22 exercises connection control in accordance with the command from the caller. In the case of the latter, the central

controller 22 automatically selects an alternative route within the IP network or a route that connects the originating telephone and the terminating telephone via a network (e.g., STM or ATM network) other than the IP
5 network and connects these terminals via the selected route.

The call data 302 is information indicative of the quality of a call currently in progress via a route within the IP network and includes delay time or loss
10 rate, day of the week and time period of the call, the IP trunk number, whether or not transfer priority has been requested, whether or not band reservation has been requested and result of evaluation as to whether voice quality is good or bad. The call data 302 is preserved
15 as voice-quality data in the system data SYDT in response to conclusion of the call.

The system data SYDT contains test-packet information 303 and voice-quality data 304. The test-packet information 303 includes (1) transmission
20 opportunity 303a, (2) packet capacity 303b of the test packet, (3) packet count 303c indicative of number of packets sent per test, and (4) test cycle 303d which specifies whether evaluation of voice quality is to be performed for every call or for every several calls.

25 The transmission opportunity 303a stipulates the method of measuring voice quality. Methods of measuring voice quality are of two types, namely (A) a post-call voice-quality measurement mode and (B) a pre-call voice-

quality measurement mode. In the post-call voice-quality measurement mode (A), voice-quality data (packet loss rate) relating to the preceding call is saved as the system data SYDT, network selection for the current
5 call is performed based upon this voice-quality data, voice-quality data is measured while the current call is in progress and this data is saved after the call in order to make the network selection for the next call. In the pre-call voice-quality measurement mode (B),
10 whether the current voice quality is good or bad is measured by transmitting a test packet before a call is established and then making the network selection based upon the result of measurement.

The voice-quality data 304 contains, for every day
15 of the week / time period of call generation, and for every call route (IP trunk number), voice-quality data such as delay time of packet loss rate, delay time or packet loss rate when transfer priority is requested, and delay time or packet loss rate when band reservation
20 is requested.

(c) IP packetizing unit

Fig. 4 is a block diagram of the IP packetizing unit 24. A switching interface 81 separates multiplexed voice data of a plurality of channels entering from the
25 side of the exchange into 64-kbps PCM voice data and outputs this voice data. Further, the switching interface 81 multiplexes PCM voice data and sends the multiplexed data to the side of the exchange.

A control unit 82 sends and receives control data to and from the central controller 22 via Ethernet, controls the evaluation of voice quality and performs other control as well. For example, if the transmission opportunity stipulates the post-call voice-quality measurement mode, the control unit 82 sends the lost packet count N_{PL} and transmitted packet count N_{PT} , which were acquired during a call, to the central controller 22 after the completion of the call, and the central controller 22 calculates the packet loss rate η using the transmitted packet count and lost packet count and saves this as loss rate in the voice-quality data 304 (see Fig. 3). If the transmission opportunity stipulates the pre-call voice-quality measurement mode, the control unit 82 sends the test-packet transmission count and the test-packet capacity per test to the control unit 82 and also instructs the control unit 82 to issue the test packets before a channel is established. The control unit 82 responds by instructing a VoIP (Voice over IP) processing unit 83 to create and send a test packet. Further, upon being notified of receipt of an ACK packet, which acknowledges the test packet, from the VoIP processing unit 83, the control unit 82 reports this to the central controller 22. The latter then calculates elapsed time from transmission of the test packet to receipt of the ACK packet as delay time and evaluates voice quality based upon the length of the delay time.

The VoIP processing unit 83 has CODECs 83a₁ - 83a_n for compressing 64-kbps voice data or restoring compressed data, IP packetizing/depacketizing units 83b₁ - 83b_n for putting compressed data into the form of IP packets (voice packets) and sending the packet to the side of the IP network, and for disassembling the voice packets received from the side of the IP network and inputting the resulting data to the CODECs, and a protocol processor 83c for executing processing in accordance with various protocols. The protocol processor 83c manages the transmitted packet count and the lost packet count and executes processing for the creation and transmission of test packets and for the receipt and reporting of ACK packets, etc. In accordance with RTP (Real Time Protocol), when packets are not received in sequential order, notification to this effect is obtained from the other party. Such notification therefore makes it possible to recognize the occurrence of a lost packet. By counting the number of notifications, the number of packets lost can be ascertained.

Ether cards 84a - 84n each incorporate a MAC controller, add a MAC header onto an IP packet and then send the IP packet from a 100Base-TX shared transmission line to the IP network via a prescribed router. Further, each Ether card removes the MAC header from a frame that enters from the IP network via a prescribed router and inputs the result to the VOIP processing unit

83.

Fig. 5A is a diagram showing the structure of a voice packet. The voice packet is composed of an IP header, a UDP header, an RTP header and voice data. The reason for using the UDP (User Datagram Protocol) and RTP is that packets must be sent continuously at regular intervals without interruption to the greatest extent possible so that voice can be reproduced in real time on the receiving side. In case of real-time voice relay, a telephone conversation can proceed even with some loss of packets and TCP (Transmission Control Protocol), therefore, is not used. The encoded voice data is sent upon being packed into the RTP user data (payload).

Fig. 5B is a diagram showing the structure of a test packet / ACK packet. The packet is composed of an IP header, a TCP header and a payload, and the type of packet, i.e., test or ACK, is specified by a control flag CF in the TCP header.

(C) Procedure for controlling telephone connection by pre-call voice-quality measurement

Fig. 6 is a diagram useful in describing a procedure for controlling telephone connection in a case where the pre-call voice-quality measurement mode is adopted. If an outgoing call is made from the telephone 10, the central controller 22 acquires from the server 70 the IP address of the IP packetizing unit in the exchange conforming to the telephone number of the call destination (step 401). Next, the central controller 22

acquires the transmission-opportunity data that has been stored in the main memory 23 (step 402) and checks to determine whether the prevailing mode in the pre-call voice-quality measurement mode (step 403). If the
5 decision is "YES", then the central controller 22 acquires the packet transmission count and the packet capacity per test packet (step 404), instructs the IP packetizing unit 24 to send a test packet and starts up a timer (step 405).

10 In response to the command to send a test packet, the IP packetizing unit 24 sends a test packet to the opposing IP packetizing unit 54 and, upon receiving the test packet, the IP packetizing unit 54 sends back an ACK packet. If the ACK packet is received, the IP
15 packetizing unit 24 so notifies the central controller 22. Accordingly, the central controller 22 stops the timer (step 406), accumulates the time from issuance of the test packet to receipt of the ACK packet (step 407), checks to determine whether the number of transmissions
20 has reached a set number (step 408) and, if the answer is "NO", causes the transmission of the next test packet and executes the processing from step 405 onward.

On the other hand, if the decision rendered at step S408 is that the number of transmissions has attained
25 the set number, the central controller 22 calculates average delay time T_A (step 409), compares the average delay time T_A with a set time T_S (step 410) and connects the originating telephone 10 and terminating telephone

60 (step 411) by executing a call set-up procedure if $T_A < T_S$ holds.

(D) Overall control of telephone connection

Fig. 7 is a status-transition diagram useful in describing overall telephone connection control, and Figs. 8A to 8J are flowcharts of a variety of processing. In Fig. 7, ST represents a symbol which indicates status, and numbers 601, 602, 603, 604 and 605 denote a status number, status description, originating telephone, switch network and trunk, respectively.

(a) Processing 1

If an outgoing call is made from telephone 10 (see Fig. 2) to telephone 60 at state 0 in Fig. 7, the central controller 22 executes processing 1 in Fig. 8A. Specifically, on the basis of the telephone number of telephone 60, the central controller 22 acquires from the server 70 the IP address of the IP packetizing unit 54 of the other exchange (step 501), then acquires the IP trunk 28b (step 502), reads the transmission-opportunity data from the system data of Fig. 3 and evaluates the transmission opportunity (step 503). There are two types of transmission opportunities, namely the post-call voice-quality measurement mode and the pre-call voice-quality measurement mode.

(1) The post-call voice-quality measurement mode is a mode for selecting a network based upon voice-quality data relating to the preceding call, measuring the voice-quality data (packet loss rate) in the course

of the current call and saving this data after the current call in order to select a network for the next call. It should be noted that the post-call voice-quality measurement mode can be implemented in the

5 following manner as well: selecting the network based upon the preceding voice-quality data, measuring delay time by sending and receiving test and ACK packets after the current call and saving the delay time in order to make the selection of a network.

10 (2) The pre-call voice-quality measurement mode is a mode for measuring the current voice quality by sending and receiving test and ACK packets before a call and then selecting a network based upon the measured voice quality.

15 If the central controller 22 determines that the transmission opportunity is indicative of the post-call voice-quality measurement mode (1) at the measurement performed at step S503, the central controller 22 executes processing 3 illustrated in Fig. 8(C) (step
20 504). If the transmission opportunity is indicative of the pre-call voice-quality measurement mode (2), then the central controller 22 executes processing 2 illustrated in Fig. 8(B) (step 505).

(b) Processing 3

25 The central controller 22 reads the preceding voice-quality data 304 (see Fig. 3), which conforms to the present day of the week / present time period of the acquired IP trunk 28b, out of the memory 23 (step 511).

Next, on the basis of the preceding voice-quality data (packet loss rate or delay time) read out, the central controller 22 evaluates the voice quality of the route through the IP network (step 512). In case of delay
5 time, the evaluation of voice quality involves rendering a "good" decision if the delay time is less than a set value (e.g., 200 ms) and a "poor" decision if the delay time is equal to or greater than the set value. The result of this evaluation is written to the call data
10 302.

(c) Processing 2

The central controller 22 reads packet capacity 303b and packet count 303c out of the main memory 23 with regard to a test packet used in order to measure
15 voice quality (step 521). Next, the central controller 22 inputs the read test-packet data to the IP packetizing unit 24 and instructs the latter to send a test packet. In response, the IP packetizing unit 24 sends a test packet using TCP (IETF RFC: 793) (step
20 522). Upon receiving an ACK packet following transmission of the test packet, the IP packetizing unit 24 so notifies the central controller 22 and the latter measures time from transmission of the test packet to receipt of the ACK packet (step 523). Measurement of
25 this elapsed time is performed by starting up a timer immediately after transmission of the test packet and stopping the timer when the ACK packet is received in response to the test packet. If the timer times out

after issuance of the test packet, then the time from issuance of the test packet to receipt of the ACK packet is adopted as time-out time. This operation is repeated a number of times equivalent to the number of
5 transmissions, the average value of measured time is obtained and, in a manner similar to that of processing 3 above, the quality is judged to be "good" if the time is less than 200 ms and "poor" if the time is equal to or greater than 200 ms (step 524). The result of this
10 decision is written to the call data 302.

(d) Processing 4

If the decision on voice quality written to the call data 302 through processing 2 or 3 is "good", the central controller 22 selects a route through the IP
15 network and calls the terminating telephone 60 (state 3) by a call set-up procedure. If the terminating telephone 60 responds by an off-hook indication, the telephones 10 and 60 are placed in a state of communication (state 4) via the IP network. If the
20 "poor" decision is rendered, the processing of Fig. 8E is executed. The foregoing is step 531 in Fig. 8D.

(e) Processing 5

If the voice quality is no good, the central controller 22 reads out the contract type 301 contained
25 in the subscriber data SBDT associated with the originating telephone 10 (step 541) and discriminates the contract type read out (step 542). There are two types of contracts, namely (A) a caller route selection

contract and (B) an automatic route selection contract.

(1) The caller route selection contract is such that if the voice quality is "poor", this is reported to the caller by the announcement unit 25 (Fig. 2) so that
5 the caller can select the route by entering a predetermined number by operating keys on the telephone 10.

(2) The automatic route selection contract is such that if the voice quality is "poor", the system selects
10 the route automatically. With automatic route selection, the caller him/herself need not select the route by operating keys.

Processing 6 in Fig. 8F is executed (step 543) in case of the contract type described in (1) above, and
15 processing 7 in Fig. 8G is executed (step 544) in case of the contract type described in (2) above.

(f) Processing 6

In case of the contract type described in (1) above, the central controller 22 connects the
20 announcement unit 25 (ANM connection) to the originating telephone 10 via the STM switch network 21 to report by voice the fact that voice quality of the route through the IP network is poor (step 551). For example, the fact that voice quality is poor is reported by the
25 following verbal message: "Poor line conditions in the present connection utilizing the IP network may cause interrupted voice". In response, a transition is made to state 1 in which the caller listens to the

announcement.

(g) Processing 7

In case of the contract type described in (2) above, the central controller 22 determines whether to
5 connect the originating telephone 10 and terminating telephone 60 via a route through the STM network or ATM network (step 561). If the connection is to be made via a route through the ATM network, the central controller 22 acquires the ATM trunk 28c (step 562). As a result,
10 state 7 is attained and the terminating telephone 60 is called. If the terminating telephone 60 responds by an off-hook indication, the telephones 10 and 60 are placed in a state of communication (state 8) via the ATM network. In case of the connection via a route through
15 the STM network, on the other hand, the central controller 22 acquires the ATM trunk 28a (step 563). As a result, state 5 is attained and the terminating telephone 60 is called. If the terminating telephone 60 responds by an off-hook indication, the telephones 10
20 and 60 are placed in a state of communication (state 6) via the STM network.

The selection of a network at step 561 is made upon taking into consideration (1) a method for selecting the ATM network if the STM network is congested, (2) a
25 method of selecting a network for which the telephone charges are low, and (3) a method of selecting a network having excellent line quality. It should be noted that a route can also be selected automatically, the route

being not only one through the STP or ATM network but also an alternative route through the IP network. In such case, taking into consideration (1) - (3) above, it would be decided to select a trunk in the following

5 order, by way of example: STM-TG (STM trunk group) ≠ ATM-TG ≠ IP-TG. If the target trunk group is congested, a trunk would be selected from the trunk group of the next rank to thereby decide the network.

(h) Processing 8 ~ 10

10 If notification is given to the effect that voice quality is poor in state 1, the caller selects (1) continue using IP network, (2) STM network, (3) ATM network, (4) IP network or (5) disconnect. The central controller 22 controls the telephone connection as
15 follows in accordance with the selection made by the caller:

(1) If a fixed period of time passes without entry of a specific number, or if the specific number is entered and its content indicates "selection to continue
20 using IP network", the central controller 22 connects the originating telephone 10 and terminating telephone 60 via the route through the IP network judged to be poor in voice quality. As a result, the terminating telephone 60 is called (state 3) and, if the terminating
25 telephone 60 responds with an off-hook indication, the telephones 10 and 60 are placed in a state of communication (state 4) via the IP network. When the call is implemented through the IP network, the central

controller 22 monitors the delay time and lost packet count of packets sent and received during the call and accumulates the monitored data in the call data 302 (see Fig. 3).

5 (2) If the content of the specific number indicates "selection of STM network", the central controller 22 acquires the ATM trunk 28a (step 571) in processing 8 shown in Fig. 8H. As a result, the terminating telephone 60 is called (state 5) and, if the
10 terminating telephone 60 responds with an off-hook indication, the telephones 10 and 60 are placed in a state of communication (state 6) via the STM network.

 (3) If the content of the specific number indicates "selection of ATM network", the central
15 controller 22 acquires the ATM trunk 28c (step 581) in processing 9 shown in Fig. 8I. As a result, the terminating telephone 60 is called (state 7) and, if the terminating telephone 60 responds with an off-hook indication, the telephones 10 and 60 are placed in a
20 state of communication (state 8) via the ATM network.

 (4) If the content of the specific number indicates "selection of IP network again", the central controller 22 executes processing 1, acquires an alternative IP trunk and makes the connection again.

25 (5) If the content of the specific number indicates "selection of disconnection", the central controller 22 disconnects the telephone.

When a call using a route through the IP network is

completed, the central controller 22 executes processing
10 shown in Fig. 8J. Specifically, the central
controller 22 calculates the packet loss rate or
measures the time delay during the call and writes the
5 same to the call data 302 (step 591). The central
controller 22 then preserves the data, which has
accumulated in the call data, as the voice-quality data
304 in the system data SYDT and subsequently utilizes
the data to evaluate voice quality (step 592). In this
10 case an arrangement can be adopted in which only one
loss rate or delay time is accumulated per route / time
period or a plurality are accumulated and voice quality
is evaluated by referring to a value obtained by
statistically processing the accumulated loss rates.
15 Examples of statistical processing averaging processing
and weighted averaging processing in which older data is
given less weight.

(D) Modifications

The foregoing embodiment relates to a case where
20 evaluation of voice quality is performed on a per-call
basis. However, an arrangement can be adopted in which
whether the evaluation is performed every call or every
several calls is set in the main memory 23 beforehand
and, if the evaluation is performed every several calls,
25 the result of evaluating voice quality is stored every
several calls and evaluation regarding a new call is
carried out by referring to the stored results of
evaluation.

The foregoing embodiment relates to a case where, if voice quality is poor, the agreement with the subscriber determines that (1) control of the telephone connection is performed in accordance with a command
5 from the user or that (2) control of the telephone connection is performed upon selecting a route automatically in accordance with the determination made by the system. However, in a case where connection of the telephone is possible only through one of these
10 methods, then the telephone is connected by that method.

Further, the foregoing embodiment relates to a case where voice quality can be evaluated in either of two modes, namely the post-call voice-quality measurement mode or the pre-call voice-quality measurement mode, and
15 the mode in which voice quality is to be evaluated is set in advance. However, an arrangement can be adopted in which evaluation is possible in one mode and the evaluation is carried out in this mode.

Thus, in accordance with the present invention as
20 described above, in a VoIP telephone connection service in which a telephone is connected via an IP network, voice quality along a route through the IP network is evaluated before a call is made. If the voice quality is good, the originating and terminating terminals are
25 connected via this route. If the voice quality is poor, then, in accordance with a command from the caller or automatically in accordance with a determination made by the system, a route other than the above-mentioned route

[e.g., an alternative route through the IP network or another route via a network (STM or ATM network) other than the IP network] is selected to control the connection of the telephone. As a result, a telephone
5 call can be implemented with an excellent voice quality.

Further, in accordance with the present invention, a test packet and an ACK packet are sent and received between an exchange on the originating side and an exchange on the terminating side via a route through the
10 IP network before a channel is established, and the voice quality of the route through the IP network is evaluated based upon the length of delay time from issuance of the test packet to receipt of the ACK packet. As a result, control of a telephone connection
15 can be performed upon measuring voice quality in real time and excellent call quality can be achieved in reliable fashion.

Further, in accordance with the present invention, voice-quality data such as packet loss rate during a
20 call is saved on a per-route basis after the call ends and the voice quality of a current call via a route through an IP network is evaluated by referring to the saved loss rate. As a result, it is unnecessary to measure voice quality before the channel is established
25 and it is therefore possible to shorten time required for a call to start.

Further, in accordance with the present invention, if voice quality of a route through the IP network is

not good, the exchange reports this to the caller by voice and the caller, by operating keys on a terminal, instructs the exchange to perform one of the following control operations: (1) to continue connection control
5 via the route for which the voice quality has been determined to be poor; (2) to perform connection control via an alternative route through the IP network; (3) to perform connection control through another route via a network other than the IP network; and (4) to perform
10 disconnect control. If this arrangement is adopted, the caller can specify control for the next telephone connection through a simple operation without being required to redial.

Further, in accordance with the present invention,
15 if voice quality of a route through the IP network is poor, an alternative route through the IP network or another route within a network (e.g., STM or ATM network) other than the IP network is selected automatically and the originating and terminating
20 terminals are connected via the selected route. As a result, greater convenience can be provided because an excellent route can be selected without requiring that the caller perform any operation.

As many apparently widely different embodiments of
25 the present invention can be made without departing from the spirit and scope thereof, it is to be understood that the invention is not limited to the specific embodiments thereof except as defined in the appended

